REMARKS

The claims pending in the application are claims 1 to 4 and 6 to 11. Claim 5 has been canceled.

Claims 1 to 4 and 6 to 11 were rejected under 35 U.S.C. §103(a) as being unpatentable over U.S. Patent No. 5,704,003 to Kleijn et al. in view of Ozawa et al., "M-LCELP Speech Coding at 4KBPS", *1994 IEEE International Conference on Acoustics, Speech, and Signal Processing*, 1994, ICASSP-94, Vol. 1, 19–22 April 1994, pp. I/269–I/272. This rejection is again respectfully traversed for the reason that the combination of Kleijn et al. and Ozawa et al. does not teach or suggest the claimed invention. It should be observed by the Examiner that the first named author of the Ozawa et al. publication is the named inventor in this patent application. Many of the arguments presented in this response and the response filed on November 12, 2003, were provided by the Applicant and, if the Examiner so desires, these arguments could be made the subject of a declaration under 37 C.F.R. §1.132 by the Applicant.

The claimed invention is directed to a speech coding apparatus for coding a speech signal at a low bit rate with high quality. The invention effectively suppresses deterioration in sound quality in terms of background noise while minimizing the calculations required. The claimed invention provides a speech coding system which succeeds in reducing the required calculations while at the same time maintaining good sound quality in terms of background noise for low bit rates. Four embodiments of the speech coding system are shown in Figures 1 to 4, respectively, and a speech decoding system is shown in Figure 5.

With reference to Figure 1 of the drawings, the invention employs a spectrum parameter calculation section which includes a spectrum parameter calculation circuit 200 for extracting spectrum parameter from a speech signal and a spectrum quantizing circuit 210 for quantizing the spectrum parameter. An adaptive codebook section includes adaptive codebook 500 and sound source quantization circuit 350. A mode discrimination circuit 370 discriminates the

mode on the basis of the past quantized gain from gain quantization circuit 366. The mode discrimination circuit 370 receives the adaptive codebook gain quantized by the gain quantization circuit 366 one subframe ahead of the current subframe and compares it to a predetermined threshold to perform voiced/unvoiced determination. When a predetermined mode is discriminated, a sound source quantization circuit 350 searches combinations of code vectors stored in a sound source code books 351 or 352, which are used to collectively quantize the amplitudes or polarities of a plurality of pulses, and a plurality of shift amounts used to temporally shift predetermined pulse positions, to select a combination of a code vector and shift amount which minimizes distortion relative to input speech. Codebook 351 is used for voiced sound and codebook 352 is used for unvoiced sound. Applicant uses separate codebooks for voiced and unvoiced speech signals so as to minimize calculations without adversely affecting sound quality in terms of background noise. A multiplexer section 400 outputs a combination of an output from the spectrum parameter calculation section, an output from the adaptive codebook section, and an output from the sound source quantization section.

The speech decoding apparatus of the invention is shown in Figure 5 and includes a demultiplexer section 510 for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information. A mode discrimination section 530 discriminates the mode on the basis of the past quantized gain of the adaptive codebook. A sound decoding section 540 reconstructs a sound source signal by generating non-zero pulses from the quantized sound source information. A speech signal is reproduced or resynthesized by passing the sound source signal through a synthesis filter 560 defined by spectrum parameters.

The second embodiment of the coding system according to the invention shown in Figure 2 differs from the first embodiment in the operation of a sound source quantization circuit 355. When voiced/unvoiced discrimination indicates an unvoiced sound, the positions are generated in advance in accordance with a

predetermined rule. A random number generating circuit 600 is used to generate a predetermined number of pulse positions which are output to the sound source quantization circuit 355. If the discrimination information indicates a voiced sound, the sound source quantization circuit 355 operates in the same manner as the sound source quantization circuit 350 in Figure 1. If, on the other hand, the information indicates an unvoiced sound, the amplitudes or polarities of pulses are collectively quantized by using a sound source codebook 352 in correspondence with the positions output from the random number generating circuit 600.

In the third embodiment of the coding system according to the invention shown in Figure 3, when voiced/unvoiced discrimination information indicates an unvoiced sound, a sound source quantization circuit 356 calculates the distortions given by equations (21) on page 31 of the specification in correspondence with all the combinations of all the code vectors in a sound source codebook 352 and the shift amounts of pulse positions, selects a plurality of combinations in the order which minimizes distortions, and outputs them to the gain quantization circuit 366. The gain quantization circuit 366 quantizes gains for a plurality of sets of outputs from the sound source quantization circuit 356 by using a codebook 380 and selects a combination of a shift amount, sound source code vector, and gain code vector which minimizes distortions.

In the fourth embodiment of the coding system according to the invention shown in Figure 4, when voiced/unvoiced discrimination information indicates an unvoiced sound, a sound source quantization circuit 357 collectively quantizes the amplitudes or polarities of pulses for the pulse positions generated by a random number generating circuit 600 by using a sound source codebook 352 and outputs all the code vectors or a plurality of code vector candidates to a gain quantization circuit 367. The gain quantization circuit 367 quantizes gains for the respective candidates output from the sound source quantization circuit 357 by using a gain codebook 380 and outputs a combination of a code vector and gain code vector which minimizes distortion.

A main feature of the present invention as recited in the claims is that a speech coding apparatus comprises a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses when an output from the discrimination section indicates a predetermined mode, and searches combinations of code vectors stored in the codebook and a plurality of shift amounts used to shift positions of the pulse so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech.

The primary reference to Kleijn et al. discloses a method of speech coding using Relaxation Code-Excited Linear Predictive (RCELP) techniques which provides a peak-to-average ratio criterion that determines whether or not time shifting of a speech residual signal should be applied within a certain sub-frame. The coders of Kleijn et al. have a characteristic feature of finding a residual signal r(n) from an input speech signal 101, and coding the residual signal r(n) by applying a time shift to the residual signal r(n) with a time warping device and delay line 107. More specifically, a time shift T, which can minimize a differential electric power between an electric power of a signal r(n−T) having a time shift T from the residual signal r(n) and that of a delayed residual signal r(n−D(n)) is firstly determined, and then a coding parameter required for the coding is extracted after applying a time shift T to the residual signal r(n).

The article by Ozawa et al. describes an M-LCELP (Multi-mode Learned Code Excited LPC) speech coder, which was developed for the North American half-rate digital cellular systems. The M-LCELP speech coder of Ozawa et al. provides multi-mode and multi-codebook coding, pitch lag differential coding with pitch tracking, a two-stage joint design regular-pulse codebook with common phase structure in voiced frames, an efficient vector quantization for LSP parameters, and an adaptive MA type comb filter to suppress excitation signal inter-harmonic noise. The M-LCELP encoder and decoder structure is shown in Fig. 1 of Ozawa et al.

The Examiner first addresses claims 1 and 6. Claim 1 is directed to the embodiments of the speech encoder shown in Figures 1 to 4, while claim 6 is directed to the combination of a coding/decoding apparatus employing the coding system of one of Figures 1 to 4 and the decoding system of Figure 5.

Claim 1 recites, *inter alia*, "a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal". This sound source quantization section comprises, "a discrimination section [**370**] for discriminating a voiced sound mode and an unvoiced sound mode on a basis of a past quantized gain of an adaptive codebook [**500**] ", and "a sound source quantization section [**350**] which has a codebook [**351, 352**] for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination circuit section, and searches combinations of code vectors stored in said codebook and a plurality of shift amounts used to shift positions of the pulses so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech". Similar limitations are recited in claims 2 and 3. Claim 6 recites "a demultiplexer section [**510**] for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information", "a mode discrimination section [**530**] for discriminating a mode by using a past quantized gain in said adaptive codebook", "a sound source signal reconstructing section [**540**] for reconstructing a sound source signal by generating non-zero pulses from the quantized sound source information when an output from said discrimination indicates a predetermined mode", and "a synthesis filter section [**560**] which is constituted by spectrum parameters and reproduces a speech signal by filtering the sound source signal".

The Examiner has stated in the paragraph at the bottom of page 2 of the Office Action that "Kleijn et al. do not specifically teach that the discriminating a voice/unvoiced mode is based on a past quantized gain of an adaptive codebook" (emphasis added). First of all, the use of the adverb "specifically" is misleading,

suggesting that Kleijn et al. might *implicitly* teach this feature, when in fact there is no suggestion whatsoever of this feature. The Examiner has further stated in the penultimate paragraph on page 3 of the Office Action that "Kleijn does not specifically teach a multiplexer for the coder or a decoder scheme with a demultiplexer and sound source reconstruction" (emphasis added). Again, the use of the adverb "specifically" is misleading, suggesting that Kleijn et al. might *implicitly* teach this feature, when in fact there is no suggestion whatsoever of this feature. In both cases, the Examiner relies on the article by Ozawa et al. to provide a teaching of the missing features and erroneously concludes that the Kleijn et al. coding system could be modified as taught by Ozawa et al. to achieve the claimed invention. The facts are to the contrary, however, since Kleijn et al. and Ozawa et al. are employing different coders based on different algorithms. Modifications of Kleijn et al. as proposed by the Examiner is not fairly taught by the references, and it is not at all clear that such modifications would result in a working coder/decoder. All the Examiner has done is to identify out of context features in Ozawa et al. and say that they could somehow be used to modify Kleijn et al. The Examiner is again reminded that the first named author of the Ozawa et al. publication is the named inventor in this patent application.

As described in the specification, a first characteristic of the present invention is that, after a speech signal is input, a voiced/unvoiced coding mode is discriminated every predetermined frame section by making use of a past quantized gain in the adoptive codebook. By this feature, it is unnecessary to newly calculate a pitch gain, etc., because the past quantized gain is used. Second, it is unnecessary to transmit mode discrimination information to the reception side. Third, therefore, the present invention can decrease a calculation amount and a transmission amount as compared with the prior art.

A second characteristic feature of the present invention is that, in a search of sound signal source (sound source codebook) consisting of the assemblage of code vectors of non-zero pulse, in a sound source quantization section, the combination of a code vector (pulse) and a shift amount for shifting a position of

the code vector (pulse) is searched on the basis of the coding mode. By the second characteristic feature which is not disclosed in either the Ozawa et al. publication or the Kleijn patent, a search coding apparatus of the present invention can increase the degree of freedom of the pulse position and improve remarkably the sound quality in a low bit rate as compared with the prior art.

On the contrary, the Ozawa et al. publication discloses a M-LCELP speech coder in which a coding mode of a predetermined frame is discriminated every predetermined frame section by calculating a pitching gain from an input speech signal. At that time, mode discrimination information must be transmitted to the reception side.

In the speech coding apparatus of the claimed invention, at the time of performing the speech coding, no coding mode information is transmitted. Therefore, in the claimed invention, it becomes possible to code a speech signal efficiently with a relatively small amount of calculation when sound source information is relatively poor and to reduce the transmitting bit rate.

On the contrary, in the prior art, coding mode information is transmitted at the time of performing the speech coding. That is, in the prior art, coding mode information is firstly obtained from a periodicity of inputted speech, then the coding mode information is transmitted. Therefore, in accordance with the coding mode information, such an operation as changing coding mode and/or excitation gain should be done.

The Examiner next groups claims 2, 5 and 7. Claim 2, like claim 1, is an independent claim directed to the several embodiments of the speech encoder shown in Figures 1 to 4, and claim 7, like claim 6, is an independent claim directed to the combination of a coding/decoding apparatus employing the coding system of one of Figures 1 to 4 and the decoding system of Figure 5. As previously mentioned, claim 5 has been canceled.

In the middle of page 4 of the Office Action, the Examiner states that "Kleijn et al. do not <u>specifically</u> teach that the discriminating a voiced/unvoiced mode is based on a past quantized gain of an adaptive code book" (emphasis

added). As already pointed out, the use of the adverb "specifically" is misleading, suggesting that Kleijn et al. might *implicitly* teach this feature, when in fact there is no suggestion whatsoever of this feature. Again, on page 5 of the Office Action, the Examiner states that "Kleijn does not specifically teach a multiplexer for the coder or a decoder scheme with a demultiplexer and sound source reconstruction" (emphasis added). Once again, the use of the adverb "specifically" is misleading, suggesting that Kleijn et al. might *implicitly* teach this feature, when in fact there is no suggestion whatsoever of this feature. As before, the Examiner relies on the article by Ozawa et al. to provide a teaching of the missing features and erroneously concludes that the Kleijn et al. coding system could be modified as taught by Ozawa et al. to achieve the claimed invention. Again, the facts are to the contrary, however, since Kleijn et al. and Ozawa et al. are employing different coders based on different algorithms. Modifications of Kleijn et al. as proposed by the Examiner is not fairly taught by the references, and it is not at all clear that such modifications would result in a working coder/decoder. All the Examiner has done is to identify out of context features in Ozawa et al. and say that they could somehow be used to modify Kleijn et al.

Next, the Examiner groups claims 3, 8 and 11. Claim 3, like claims 1 and 2, is an independent claim directed to the embodiments of the speech encoder shown in Figures 1 to 4. Claim 8 is also an independent claim directed to a speech coding apparatus, and claim 11 is a dependent claim dependent on claim 8 and recites *inter alia* that, when mode discrimination indicates a predetermined mode, the sound quantization means selects a plurality of combinations from combinations of all code vectors in the codebook and shift amounts for pulse positions in an order in which a predetermined distortion amount is minimized.

Claim 8 is directed to a speech coding apparatus which comprises, *inter alia*, "mode discrimination means [370] for receiving a past quantized adaptive codebook gain and performing mode discrimination associated with a voiced/unvoiced mode by comparing the gain with a predetermined threshold" and "sound source quantization means [350] for quantizing a sound source signal of

the speech signal by using the spectrum parameter and outputting the signal, and searching combinations of code vectors stored in a codebook for collectively quantizing amplitudes or polarities of a plurality of pulses in a predetermined mode and a plurality of shift amounts used to temporally shift a predetermined pulse position so as to select a combination of an index of a code vector and a shift amount which minimizes distortion relative to input speech".

In the paragraph bridging pages 5 and 6 of the Office Action, the Examiner states that "Kleijn et al. do not specifically teach that the discriminating a voiced/unvoiced mode is based on a past quantized gain of an adaptive codebook" (emphasis added), and at the top of page 7, the Examiner states that "Kleijn does not specifically teach a multiplexer for the coder or a decoder scheme with demultiplexer and sound source reconstructions" (emphasis added). Once again, the use of the adverb "specifically" is misleading, suggesting that Kleijn et al. might *implicitly* teach this feature, when in fact there is no suggestion whatsoever of this feature. As before, the Examiner relies on the article by Ozawa et al. to provide a teaching of the missing features and erroneously concludes that the Kleijn et al. coding system could be modified as taught by Ozawa et al. to achieve the claimed invention. Again, the facts are to the contrary, however, since Kleijn et al. and Ozawa et al. are employing different coders based on different algorithms. Modifications of Kleijn et al. as proposed by the Examiner is not fairly taught by the references, and it is not at all clear that such modifications would result in a working coder/decoder. All the Examiner has done is to identify out of context features in Ozawa et al. and say that they could somehow be used to modify Kleijn et al.

Regarding claim 4, this claim recites, "a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal". The sound source quantization section comprises, "a discrimination section **[370]** for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain of an adaptive codebook **[500]**", and "a sound source quantization section **[350]**

which has a codebook [351, 352] for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on output from said discrimination section, and a gain codebook [380] for quantizing gains, and outputs a combination of a code vector and gain code vector which minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule".

In the penultimate paragraph on page 7 of the Office Action, the Examiner states that "Kleijn et al. do not specifically teach that the discriminating a voiced/unvoiced mode is based on a past quantized gain of an adaptive codebook" (emphasis added), and in the fourth paragraph from the bottom on page 8, the Examiner states that "Kleijn does not specifically teach a multiplexer for the coder or a decoder scheme with a demultiplexer and sound source reconstruction" (emphasis added). Once again, the use of the adverb "specifically" is misleading, suggesting that Kleijn et al. might *implicitly* teach this feature, when in fact there is no suggestion whatsoever of this feature. As before, the Examiner relies on the article by Ozawa et al. to provide a teaching of the missing features and erroneously concludes that the Kleijn et al. coding system could be modified as taught by Ozawa et al. to achieve the claimed invention. Again, the facts are to the contrary, however, since Kleijn et al. and Ozawa et al. are employing different coders based on different algorithms. Modifications of Kleijn et al. as proposed by the Examiner is not fairly taught by the references, and it is not at all clear that such modifications would result in a working coder/decoder. All the Examiner has done is to identify out of context features in Ozawa et al. and say that they could somehow be used to modify Kleijn et al.

Finally, the Examiner groups claims 9 and 10, both of which are dependent on claim 8. Claim 9 recites that the sound source quantization means uses a position generated according to a predetermined rule as a pulse position when mode discrimination indicates a predetermined mode. Claim 10 recites that, when mode discrimination indicates a predetermined mode, a predetermined number of

pulse positions are generated by random number generating means and output to the sound source quantization means. This is shown in the embodiments of Figures 2 and 4. The Examiner cites col. 7, lines 2–26, of Kleijn et al. for a disclosure of the features recited in claims 9 and 10; however, the cited passage does not in fact support the Examiner. What Kleijn et al. describe is calculation of ov the value of G_{opt} and then ascertaining whether or not the calculated value is greater than a first specified threshold value. This is not what is being claimed.

In his response to the arguments filed on November 12, 2003, the Examiner states that he "recognizes that obviousness can *only* be established by combining or modifying the teachings of the prior art to produce the claimed invention where there is some *teaching*, *suggestion*, or *motivation to do so found in the references themselves* or *in the knowledge generally available to one of ordinary skill in the art*" (emphasis added). Having said that, the Examiner abandons all effort to demonstrate that there is "some *teaching*, *suggestion*, or *motivation to do so found in the references themselves*", since there is in fact none, but relies instead on some notion of "*knowledge generally available to one of ordinary skill in the art*", without demonstrable evidence supporting his position. If the Examiner knows of such evidence in the form of a publication or issued patent, he is requested to cite it. If the Examiner is relying on his own knowledge, then he is requested to provide an affidavit stating the nature of his knowledge, when he acquired it, and under what circumstances. Failing either the citation of such a publication or issued patent or the submission of a probative affidavit by the Examiner, the rejection of the claims can only be viewed as a subjective application of the standards under Section 103 based on hindsight, whereas the appropriate standard is an objective standard. The rejection should therefore be withdrawn.

In view of the foregoing, it is respectfully requested that the application be reconsidered, that claims 1 to 4 and 6 to 11 be allowed, and that the application be passed to issue.

Should the Examiner find the application to be other than in condition for

allowance, the Examiner is requested to contact the undersigned at the local telephone number listed below to discuss any other changes deemed necessary in a telephonic or personal interview.

A provisional petition is hereby made for any extension of time necessary for the continued pendency during the life of this application. Please charge any fees for such provisional petition and any deficiencies in fees and credit any overpayment of fees to Attorney's Deposit Account No. 50-2041.

Respectfully submitted,

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